INTERNET TIME MULTIPLEXED CIRCUIT CONNECTION FOR WIRE SPEED CONNECTION AKIN TO PSTN SWITCHED CIRCUIT CONNECTION SUITABLE FOR MULTIMEDIA/VOICE/FAX/REALTIME APPLICATIONS

At present to facilitate multimedia/voice/fax/realtime applications on the Internet requires the IP packets to be given priority over other packets by methods such as RSVP/Tag Switching to ensure Quality of Service.

Here is presented a method whereby an Internet Time multiplexed Circuit Connection is established enabling data communications at both ends in exactly the same way as in the case where the Internet Connection so established is a PSTN switched circuit connection, at wire speed with same transmissions quality. Data is transmitted without IP packetising, session & communications protocols are handled by & between both end users.

A Worldwide Connections Manager arranges at predetermined periods all selected nodes between source & destination to automatically switch incoming signals to next node at wire speed without buffering delay/route computation delay, thus establishes a Time Multiplexed Circuit Connection for the durations of the predetermined periods, as in the case where a simplex PSTN dedicated circuit connection has been established & individual digital signal bits could be transmitted at any time & received instantaneously. For Duplex communication requirements, 2 unidirectional Virtual dedicated circuit connections are set up, one in each directions (& could be of different bandwidths), preferably along the route in both directions with prearranged automatic switchings of transmissions at the same predetermined time periods to have similar transmission line dependant characteristics. Half Duplex

will have 2 unidirectional Time Multiplexed Circuit Connections with only one of them needs to be maintained at any one time.The caller & receiving side nodes/ITSP access the Worldwide Connections Manager for simplex/half-duplex/full duplex Time Multiplexed Circuit Connections initialisations, monitoring & releases.

A number of individual bit transmissions (continuously) from various sources could be conveniently interleaved time multiplexed at source side node/ITSP , instantaneously received (as in PSTN transmissions) & demultiplexed at destination side node/ITSP (converted to analog if required) thereby forwarded to various destination receivers. Worldwide Connections Manager may arrange so that the routers at the nodes automatically switches various incoming Time Multiplexed Circuit Connection transmissions from several neighbouring nodes contiguosly time multiplexed (ie the various predetermined time periods bandwidths coincide as a larger continuous predetermined time periods transmissions) onto next common node hops by intelligent schedulings of predetermined time periods bandwidth allocations during initialisations of Virtual dedicated circuit connections (so that predetermined time frames allocated to various sources merge seamlessly interleaved time multiplexed into bigger continuous blocks for onward next node hops).

In practise, for a single Voice telephony, the 4 KHz analog signals at say 1/10 second intervals are first converted to 6.4 Kbits/sec digital signals using PCM (sampling at twice the analog frequency, & uncompressed/compressed as required) and then time division concentrated/multiplexed along with regular IP packet traffics by the caller side node/ITSP (as 1/millionth of a second duration digital transmissions) onto say the 6.4 Gigabits/sec transmission link (ie equates to the communications rate of the lowest link along the nodes) to the next node (which in turn automatically switches the transmissions onto the next node, & so forth) along the Time Multiplexed

Circuit Connection as raw unpacketised digital signals of 1/millionth of a second duration at the precise prearranged 1/millionth of a second time period where the Time Multiplexed Circuit Connection between Source & destination is established. The 1/millionth of a second digital signals received at receiver side node/ITSP is then converted back into a 1/10 second 4 KHz analog signals to be forwarded to recipient. The next 1/10 second interval 4 KHz analog signal from caller is again digitised, transmitted along the Time Multiplexed Circuit Connection at the next 1/millionth of a second period (occuring at intervals of 1/10 second) , reconverted back into a 1/10 second interval 4 KHz analog signals at receiver side node/ITSP, & received at recipient as a continuous series of 1/10 second speech. No network delays due to transmission on the Internet will be encountered as in present Voice IP packet transmissions. The total delays between the speech at caller & reception at recipient here is 1/10 second plus 2 x analog/digital conversion delays (~30 msec), ie total fixed delay of 0.16 second regardless of distance (ignoring the time for the uncompressed signals to travel along the transmission medium).

A thousand originating calls from say London could be interleaved time multiplexed at source side node/ITSP as a single 1/thousandth of a second duration digital transmissions onto the 6.4 Gigabits/sec transmission link at the precise prearranged 1/thousandth of a second time periods where the Time Multiplexed Circuit Connection between source & destination is established, received at destination node/ITSP, demultiplexed & converted back into one thousand 1/10 second duration 4 KHz analog speech signals to be forwarded along dialled up PSTN lines to receiving telephones.

Not all links along the selected nodes communicates at the same rate. Say the single Voice signals above traverse nodes A, B, C, D and the links A-B communicates at 6.4Megabits/sec, B-C at 6.4 Gigabits/sec & C-D at 6.4 Megabits/sec, node A will be prearranged to automatically switch incoming transmission (PCM digitised 1/10 second 4KHz analog signals) onto node B starting at Time 0 for 1/thousandth of a second duration. The 1/thousandth of a second duration transmission arriving at node

B between Time 0 & Time 0+1/thousandth second is automatically switched at Time 0 onto node C for same 1/thousandth of a second duration (ie interleaved time multiplexed, regularly allocated one slot for every 1,000th slots in the 6.4 Gigabits/sec transmission bandwidth during the 1/thousandth of a second prearranged time period).The 1/thousandth of a second duration transmission arriving at node C between Time 0 & Time 0+1/thousandth second will be automatically demultiplexed to obtain the PCM Voice signals to be automatically switched onto destination node D arriving during Time 0 & Time 0+1/thousandth second (the 1/thousandth of a second time period here is the time period required for 1/10 second interval Voice speech to be transmitted along the link of the lowest communication rate, ie link A-B / C-D). It may seem to be wasteful of allocated time multiplexed Bandwidth at link B-C, this is overcomed by intelligently arranging for similar 1/Millionth of a second duration incoming Time Multiplexed Circuit Connection transmissions from other nodes neighbouring node B to arrive at node B at the right time slots (less time critical arriving transmissions may be interleaved time slot shifted) to be interleaved time slot multiplexed into the 1/thousandth of a second bandwidth for common next node hop onto node C.
[Alternatively without hardware demultiplexing of 6.4 Gigabits/sec incoming transmission at node C, node B could be prearranged to automatically switch transmission onwards to node C a thousand times (each time for 1/thousand millionth second duration, ie corresponding to transmission time for a single time multiplexed slot) during Time 0 & Time 0+1/thousandth second. Node C transmits each PCM signals contained in the 1,000 time multiplexed slots, between Time 0 & Time 0+1/thousandth second automatically onto node D along the slower 6.4 Megabits/sec link, having first buffered the incoming time multiplexed Voice slots to adjust for the different communication rates of the links (without incurring signal transmission delays here as the signals are buffered for onwards slower transmission) .]

The nodes are synchronised to each other for transmissions. The Time 0 referred to above ignores the time for signals to travel along the transmission medium. In reality,

the Time 0 at nodes A, B, C & D are each adjusted to take into account the time it take for signals to arrive travelling along the immediately preceding links.